PATENT COOPERATION TREATY

PCT

INTERNATIONAL SEARCH REPORT

(PCT Article 18 and Rules 43 and 44)

Applicant's or agent's file reference	**FOR FURTHER ACTION** see Notification of Transmittal of International Search Report (Form PCT/ISA/220) as well as, where applicable, item 5 below.	
RCA 88797		
International application No.	International filing date *(day/month/year)*	(Earliest) Priority Date *(day/month/year)*
PCT/US 99/ 01631	27/01/1999	27/01/1998
Applicant		
THOMSON CONSUMER ELECTRONICS, INC. et al.		

This International Search Report has been prepared by this International Searching Authority and is transmitted to the applicant according to Article 18. A copy is being transmitted to the International Bureau.

This International Search Report consists of a total of 3 sheets.

[X] It is also accompanied by a copy of each prior art document cited in this report.

1. **Basis of the report**

   a. With regard to the **language,** the international search was carried out on the basis of the international application in the language in which it was filed, unless otherwise indicated under this item.

   [ ] the international search was carried out on the basis of a translation of the international application furnished to this Authority (Rule 23.1(b)).

   b. With regard to any **nucleotide and/or amino acid sequence** disclosed in the international application, the international search was carried out on the basis of the sequence listing :

   [ ] contained in the international application in written form.

   [ ] filed together with the international application in computer readable form.

   [ ] furnished subsequently to this Authority in written form.

   [ ] furnished subsequently to this Authority in computer readble form.

   [ ] the statement that the subsequently furnished written sequence listing does not go beyond the disclosure in the international application as filed has been furnished.

   [ ] the statement that the information recorded in computer readable form is identical to the written sequence listing has been furnished

2. [ ] **Certain claims were found unsearchable** (See Box I).

3. [ ] **Unity of invention is lacking** (see Box II).

4. With regard to the **title,**

   [X] the text is approved as submitted by the applicant.

   [ ] the text has been established by this Authority to read as follows:

5. With regard to the **abstract,**

   [X] the text is approved as submitted by the applicant.

   [ ] the text has been established, according to Rule 38.2(b), by this Authority as it appears in Box III. The applicant may, within one month from the date of mailing of this international search report, submit comments to this Authority.

6. The figure of the **drawings** to be published with the abstract is Figure No. 4

   [X] as suggested by the applicant.

   [ ] because the applicant failed to suggest a figure.

   [ ] because this figure better characterizes the invention.

   [ ] None of the figures.

PATENT COOPERATION TREATY

From the INTERNATIONAL BUREAU

PCT

NOTIFICATION OF ELECTION

(PCT Rule 61.2)

To:

Assistant Commissioner for Patents
United States Patent and Trademark Office
Box PCT
Washington, D.C.20231
ÉTATS-UNIS D'AMÉRIQUE

in its capacity as elected Office

Date of mailing (day/month/year) 02 September 1999 (02.09.99)	
International application No. PCT/US99/01631	**Applicant's or agent's file reference** RCA 88797
International filing date (day/month/year) 27 January 1999 (27.01.99)	**Priority date** (day/month/year) 27 January 1998 (27.01.98)
Applicant RHODES, Robert, Andrew et al	

1. The designated Office is hereby notified of its election made:

[X] in the demand filed with the International Preliminary Examining Authority on:

04 August 1999 (04.08.99)

[ ] in a notice effecting later election filed with the International Bureau on:

2. The election [X] was

[ ] was not

made before the expiration of 19 months from the priority date or, where Rule 32 applies, within the time limit under Rule 32.2(b).

The International Bureau of WIPO **34, chemin des Colombettes** **1211 Geneva 20, Switzerland**	Authorized officer Jean-Marie McAdams
Facsimile No.: (41-22) 740.14.35	Telephone No.: (41-22) 338.83.38

PATENT COOPERATION TREATY

From the
INTERNATIONAL PRELIMINARY EXAMINING AUTHORITY

To:

THOMSON MULTIMEDIA LICENSING INC.
PO Box 5312
PRINCETON, NEW JERSEY 08543
ETATS-UNIS D'AMERIQUE

PCT

NOTIFICATION OF TRANSMITTAL OF THE INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Rule 71.1)

Date of mailing *(day/month/year)*	29.05.2000
Applicant's or agent's file reference: RCA 88797	**IMPORTANT NOTIFICATION**

International application No.	International filing date *(day/month/year)*	Priority date *(day/month/year)*
PCT/US99/01631	27/01/1999	27/01/1998

Applicant
THOMSON CONSUMER ELECTRONICS, INC. et al.

1. The applicant is hereby notified that this International Preliminary Examining Authority transmits herewith the international preliminary examination report and its annexes, if any, established on the international application.

2. A copy of the report and its annexes, if any, is being transmitted to the International Bureau for communication to all the elected Offices.

3. Where required by any of the elected Offices, the International Bureau will prepare an English translation of the report (but not of any annexes) and will transmit such translation to those Offices.

4. **REMINDER**

The applicant must enter the national phase before each elected Office by performing certain acts (filing translations and paying national fees) within 30 months from the priority date (or later in some Offices) (Article 39(1)) (see also the reminder sent by the International Bureau with Form PCT/IB/301).

Where a translation of the international application must be furnished to an elected Office, that translation must contain a translation of any annexes to the international preliminary examination report. It is the applicant's responsibility to prepare and furnish such translation directly to each elected Office concerned.

For further details on the applicable time limits and requirements of the elected Offices, see Volume II of the PCT Applicant's Guide.

Name and mailing address of the IPEA/	Authorized officer
European Patent Office D-80298 Munich Tel. +49 89 2399 - 0 Tx: 523656 epmu d Fax: +49 89 2399 - 4465	Pelatti, V Tel.+49 89 2399-7309

Form PCT/IPEA/416 (July 1992)

PATENT COOPERATION TREATY

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INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Article 36 and Rule 70)

Applicant's or agent's file reference	FOR FURTHER ACTION	See Notification of Transmittal of International Preliminary Examination Report (Form PCT/IPEA/416)
RCA 88797		
International application No.	International filing date *(day/month/year)*	Priority date *(day/month/year)*
PCT/US99/01631	27/01/1999	27/01/1998
International Patent Classification (IPC) or national classification and IPC H04M7/00		
Applicant THOMSON CONSUMER ELECTRONICS, INC. et al.		

1. This international preliminary examination report has been prepared by this International Preliminary Examining Authority and is transmitted to the applicant according to Article 36.

2. This REPORT consists of a total of 6 sheets, including this cover sheet.

   ☒ This report is also accompanied by ANNEXES, i.e. sheets of the description, claims and/or drawings which have been amended and are the basis for this report and/or sheets containing rectifications made before this Authority (see Rule 70.16 and Section 607 of the Administrative Instructions under the PCT).

   These annexes consist of a total of 5 sheets.

3. This report contains indications relating to the following items:

   - I ☒ Basis of the report
   - II ☐ Priority
   - III ☐ Non-establishment of opinion with regard to novelty, inventive step and industrial applicability
   - IV ☐ Lack of unity of invention
   - V ☒ Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations suporting such statement
   - VI ☒ Certain documents cited
   - VII ☒ Certain defects in the international application
   - VIII ☒ Certain observations on the international application

Date of submission of the demand	Date of completion of this report
04/08/1999	29.05.2000
Name and mailing address of the international preliminary examining authority: European Patent Office D-80298 Munich Tel. +49 89 2399 - 0 Tx: 523656 epmu d Fax: +49 89 2399 - 4465	Authorized officer Teiwes, J Telephone No. +49 89 2399 7504

Form PCT/IPEA/409 (cover sheet) (January 1994)

INTERNATIONAL PRELIMINARY EXAMINATION REPORT

International application No. PCT/US99/01631

I. Basis of the report

1. This report has been drawn on the basis of (*substitute sheets which have been furnished to the receiving Office in response to an invitation under Article 14 are referred to in this report as "originally filed" and are not annexed to the report since they do not contain amendments.*):

Description, pages:

1,2,5-12	as originally filed	
3,4,4a	with telefax of	15/05/2000

Claims, No.:

1-8	with telefax of	15/05/2000

Drawings, sheets:

1/6-6/6	as originally filed

2. The amendments have resulted in the cancellation of:

- ☐ the description, pages:
- ☐ the claims, Nos.:
- ☐ the drawings, sheets:

3. ☐ This report has been established as if (some of) the amendments had not been made, since they have been considered to go beyond the disclosure as filed (Rule 70.2(c)):

4. Additional observations, if necessary:

V. Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement

1. Statement

Novelty (N)	Yes:	Claims	1-8
	No:	Claims	
Inventive step (IS)	Yes:	Claims	
	No:	Claims	1-8
Industrial applicability (IA)	Yes:	Claims	1-8
	No:	Claims	

2. Citations and explanations

see separate sheet

VI. Certain documents cited

1. Certain published documents (Rule 70.10)

and / or

2. Non-written disclosures (Rule 70.9)

see separate sheet

VII. Certain defects in the international application

The following defects in the form or contents of the international application have been noted:

see separate sheet

VIII. Certain observations on the international application

The following observations on the clarity of the claims, description, and drawings or on the question whether the claims are fully supported by the description, are made:

see separate sheet

Re Item V

Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement

1 The following document will be referenced:

D1: WO 97 29581 A (LINK WORLDWIDE INC I) 14 August 1997
D2: WO 98 11703 A (SOLOMON YORAM ;SOLRAM ELECTRONICS LTD (IL)) 19 March 1998

2 The present application does not meet the requirements of Articles 33(1) and (3) PCT, because the subject-matter of claim 1 is not inventive. Claim 1 is therefore not allowable.

2.1 The present formulation of independent method claim 1 is such that most of its subject matter can be read onto prior art document D1, which relates to a voice internet transmission system for processing voice calls over Internet (p.1, l.5-13).

In particular, document D1 discloses the step of receiving a signal from a cable network (p.12, l.18-22; p.13, l.26-29; fig.2, obj.31, "cable") , the signal representing data of a voice call, the signal being both modulated in a first format and compressed to match a format of the cable network (p.15, l.30-35; fig.2, obj.32,34).
It further discloses the step of demodulating said signal modulated in said first format and decompressing said signal (p.15, l.30 - p.16, l.4; fig.2, obj.34).
D1 further discloses the step of compressing the signal into a format of a home environment and modulating the compressed signal into a second format (fig.2, obj.31 "cellular, radio, cable,..."). Even if *compressing the signal into a format of the home environment and modulating the compressed signal into a second format* is not explicitly mentioned in D1, these method steps are implicitly disclosed by any cellular, cable or radio system used to provide a link from the originating/receiving phone to the Internet (fig.2, obj.31).
D1 further discloses the step of wirelessly transmitting the signal compressed in the format of the home environment and modulated in the second format to a wireless device (fig.2, obj.31, 38).

It further discloses the step of demodulating and decompressing the signal in the wireless device (fig.2, obj.38).

The subject matter of claim 1 differs from D1 in that the signal received from the cable network represent Internet protocol data packets of the voice call. But processing the Internet data packets at different sites, i.e. before or after sending them via the cable, does not influence the processing method itself and is merely considered as being a measure of a common technical design not involving any inventive merit.

2.2 The additional features of dependent claim 2-4 are common design measures. Hence, these features do not add anything inventive to claim 1.

3 The features of independent system claim 5 correspond to the already discussed features of independent method claim 1 which are disclosed by D1. Hence, claim 5, does not involve anything inventive.

3.1 The features of dependent claims 6-7 correspond to the already discussed features of dependent claims 2-3 (see para. 2.2). Hence, claim 6-7 not add anything inventive to claim 5.

3.2 Claim 7 is identical with claim 4 (see para. 2.2; see **Item VIII** 1).

Re Item VI
Certain documents cited

Certain published documents (Rule 70.10 PCT)

Application No Patent No	Publication date *(day/month/year)*	Filing date *(day/month/year)*	Priority date *(valid claim)* *(day/month/year)*
WO 98 11703	19.3.98	26.8.97	16.9.96

The priority document of the application was not at available at the time of the preliminary examination. It is therefore assumed that the priority of the application is valid in consequence of which **document D2=WO 98 11703** is not considered as prior

art. Said assumption will be invalid if it turns out that the priority of the application is not valid and then the document could become relevant in a later regional or national examination procedure.

Re Item VII

Certain defects in the international application

1 In order to meet the requirements of Rule 6.3(b) PCT the independent claims should have been cast in the two-part form, with those features which in combination are disclosed by document D1 should have been placed in a preamble Rule 6.3(b)(i) PCT and with the remaining features being included in a characterising part, Rule 6.3(b)(ii) PCT.

2 Reference signs placed in parentheses should have been inserted into the claims to increase their intelligibility (see Rule 6.2(b) PCT). This applies to both the preamble and the characterising portion.

Re Item VIII

Certain observations on the international application

1 Claim 8 is identical with claim 4. Hence, claim 8 should have been deleted for conciseness and clarity reasons (Article 6 PCT).
Or claim 8 should have been amended by replacing the wording "The method of claim 1,..." by "The system of claim 5,..." because it might have been the intention of the Applicant to draft a corresponding system claim to method claim 4.

2 The use of the term "the spirit of the invention" (p.12, l.9, l.27) casts doubt on the scope of the claims and should have been deleted (see PCT-Guidelines chapter III 4.3a).

a dial-up connection call, a phone that has been called won't ring unless the Internet connection is already established for this phone. For a direct connection call, a phone would ring like a normal telephone.

The are many advantages to IP/Internet telephony. One such advantage is reduced cost as described above. A low bit rate audio codec embedded in the IP voice device enables voice calls over a 28.8 Kbps modem. For a small reduction in voice quality, a person's monthly phone bill will be greatly reduced. If IP voice device used together with a cable modem, the private service network plus high bandwidth of the cable modem will provide very good sound quality. Even if the voice quality provided by a IP/internet voice device is unsuitable for all phone communications, a IP/internet voice device may be useful as a second-line residential phone. Also, the H.323 standard supports several well defined conference modes and, therefore, IP voice device is able to be used for multi-point conference calls. A "Web" dial-in service is advantageous for technical or customer support lines because, for example, an Internet address of a company's IP voice device can be embedded in the company's Web page and customers can then call the company simply by "clicking on" that Internet address. The cost associated with toll-free ("800" number) telephone numbers will be reduced as a result.

In addition, MSOs (cable television system operators) have recently become interested in adding inexpensive telephony services using a combination of an MSO's private HFC (Hybrid Fiber Coax) network and the public Internet. Voice signals are converted to digital values and transported across the networks using various established and proposed Internet protocols as IP (internet protocol) packets.

Reference D1 (WO 97 29581 A) discloses a transmission system which enables users to have a voice conversation via the internet. D1 utilizes the PSTN to connect with an internet service provider. The system disclosed by D1 includes transmitting a telephone call to the PSTN and an originating voice engine which compresses the signal for transmission over the internet. The

signal is then transmitted over the internet to a receiving voice engine. The receiving voice engine decompresses and demodulates the received signal and provides the signal to the PSTN (31). The PSTN compresses the signal into a format for transmission therealong for receipt by a receiving telephone. This system compresses and decompresses a signal into a format suitable for transmission by the PSTN. Furthermore, the compression and decompression of the signal is performed at the PSTN, remotely from the ends of the established communication channel. Thus, although this system eliminates most long distance charges associated with a voice call, there are still local charges associated therewith and possibly long distance charges on either side of the communication channel associated with contacting the internet service provider.

Reference D2 WO 98 11703 was published 19 March 1998.

However, there are also problems associated with existing IP/Internet telephony systems. For example, the above-described systems involve some combination of additional or revised POTS (Plain Old Telephone System) wiring, additional or revised cable network wiring, or additional network interface boxes. In addition, any connection which replaces a PSTN (public switched telephone network) service (such as reuse of the existing POTS wiring within the home to replace PSTN services with HFC telephony services) may be required to supply so called "life-line" services. Some of these options require professional installation which may be costly, time consuming, and inconvenient for the user.

SUMMARY OF THE INVENTION

The invention resides, in part, in recognition of the above-described problems and, in part, in providing a system and method for solving these problems. In particular, the inventors recognize that the described problems are solved by providing a voice call over an internet connection by receiving a signal from a cable network. The signal represents internet protocol data packets of the voice call and is both modulated in a first format and compressed to match a format of the cable network. The signal is demodulated and decompressed. The signal is next compressed into a format of a home environment, modulated into a

second format and wirelessly transmitted to a wireless device. The signal is then demodulated and decompressed in the wireless device. The inventors also provide a system including the elements necessary to carry out this method.

An aspect of the present invention involves providing an internet telephony system using a wireless connection such as via the unregulated 900 MHz cordless phone spectrum or other spectrum allocated for wireless communications to provide an RF link between an IP connection device, a network interface box or set-top box; and one or more wireless handsets. A processing/control element in the network interface box would run the required IP protocols to establish and manage call set-up and teardown (currently defined within ITU-T H.323), translate the digital voice signal between IP and the local RF link protocol, and provide the RF base station function for the handset(s). Each handset would incorporate the other end of the RF link, and A/D and D/A functions to convert the voice signal to and from digital packets, and potentially apply some compression algorithm to improve bandwidth utilization. In a handset design which does not incorporate enough processing power to perform the compression function, this function could potentially reside in the network interface box.

Another aspect of the present invention involves a mechanism to establish a wireless interface to a telephone handset through a settop box that is tied into a cable network such as a hybrid coaxial cable network.

Another aspect of the present invention involves using a standard protocol such as the Internet Protocol to maintain a digital connection into a cable network while using an RF link to transmit compressed voice/data information between a telephone device such as a telephone handset and an interface unit such as a settop box.

CLAIMS

1. A method for processing a voice call over an internet, comprising the steps of:

receiving a signal from a cable network, the signal representing internet protocol data packets of the voice call and being both modulated in a first format and compressed to match a format of the cable network;

demodulating the signal modulated in the first format;

decompressing the signal;

compressing the signal into a format of a home environment;

modulating the compressed signal into a second format;

wirelessly transmitting the signal compressed in the format of the home environment and modulated in the second format to a wireless device; and

demodulating and decompressing the signal in the wireless device.

2. The method of claim 1, wherein the first format is H.323 compliant.

3. The method of claim 1, wherein the first format comprises a same modulation scheme as the second format.

4. The method of claim 1, wherein the first format comprises a different modulation scheme as the second format.

5. A system for processing a voice call over an internet, comprising:

means for receiving, demodulating and decompressing a signal representing internet protocol data packets of the voice call, the signal being received from a cable network, modulated in a first format and compressed in a format of the cable network;

means for modulating the signal into a second format and compressing the signal into a format of a home environment for wireless transmission of the modulated and compressed signal; and

a wireless device including means for demodulating and decompressing the signal for completion of the voice call.

6. The system of claim 5, wherein the first format is H.323 compliant.

7. The system of claim 5, wherein the first format comprises a same modulation scheme as the second format.

8. The method of claim 1, wherein the first format comprises a different modulation scheme as the second format.

PCT WORLD INTELLECTUAL PROPERTY ORGANIZATION
International Bureau

INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 6 : H04M 7/00 | A1

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(30) Priority Data:
60/072,649 27 January 1998 (27.01.98) US

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(72) Inventors; and

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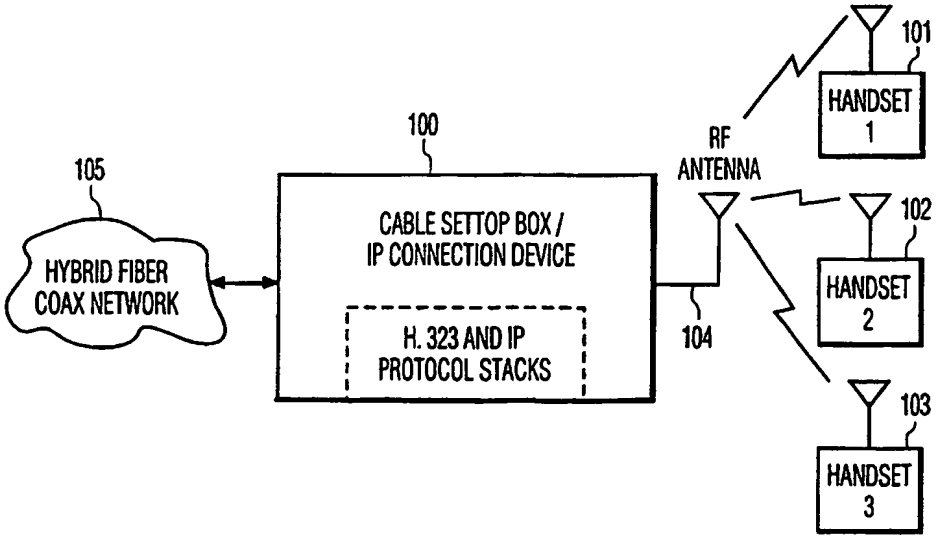
(74) Agents: TRIPOLI, Joseph, S. et al.; Thomson Multimedia Licensing Incorporated, P.O. Box 5312, Princeton, NJ 08540 (US).

(81) Designated States: AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, CA, CH, CN, CU, CZ, DE, DK, EE, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MD, MG, MK, MN, MW, MX, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, SL, TJ, TM, TR, TT, UA, UG, US, UZ, VN, YU, ZW, ARIPO patent (GH, GM, KE, LS, MW, SD, SZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GW, ML, MR, NE, SN, TD, TG).

Published
With international search report.
Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.

(54) Title: SYSTEM AND METHOD FOR PROVIDING IP/INTERNET TELEPHONY



(57) Abstract

A system and method are described providing a wireless interface to an Internet Protocol (IP)/Internet telephony system. A wireless connection via the unregulated 900 MHz cordless phone spectrum or other spectrum allocated for wireless communications provides an RF link between an IP connection device, a network interface box or set–top box, and one or more wirelss handsets. A processing/control element in the network interface box runs the required IP protocols to establish and manage call set–up and teardown, translate the digital voice signal between IP and the local RF link protocol, and provide the RF base station function for the handset(s). Each handset would incorporate the other end of the RF link. The wireless interface to a telephone handset may be through a settop box that is tied into a cable network such as a hybrid coaxial cable network. A protocol such as the Internet Protocol may be used to maintain a digital connection into a cable network while using an RF link to transmit compressed voice/data information between a telephone device such as a telephone handset and an interface unit such as a settop box.

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SYSTEM AND METHOD FOR PROVIDING IP/INTERNET TELEPHONY

FIELD OF THE INVENTION

The present invention generally relates to a system and method for providing internet telephony. In particular, the present invention relates to a system and method of providing a wireless internet telephone system over either a regular dial up telephone or a cable network.

BACKGROUND

One of the primary reasons for interest in offering Internet Protocol (IP)/internet telephony services is the pricing structures currently in place for the data service, and voice service offered by telephone operators. Long distance voice service can be thought of as "demand data" service, where the user pays a premium for the instantaneous access to a 64 Kbps channel (voice grade channel in the US). Widely publicized, promotional type pricing for this service is on the order of $.10 a minute. By contrast, data service offered by telephone operators, such as that offered for a T-1 connection (24 voice quality data lines, for a 1.544 Mbps connection) is priced at approximately $1000/month, which works out to $.001 a minute per voice line. In the very near future, cable operators will place extreme pressure on even the data service rates for telephone operators, as cable modem will allow cable operators to offer hundreds of Kbps effective throughput for approximately $50/month.

The basic idea of IP/internet voice telephony is to digitize your voice as you talk on the phone and send the digitized data as IP packets to the Internet. An IP voice device can be embedded within an Internet connection device such as a modem, a set-top-box, or a computer. It can be also built as a stand alone product. The stand alone IP voice device, for example, may provide an Ethernet jack which can be connected with an Internet connection device and other LAN devices. The IP voice device may also include interfaces to connect regular phone handsets. The quality of speech heard through a normal telephone line requires 64kbits/s bandwidth. However, most current internet connections have less bandwidth, such as 28.8 Kbps, or 56 Kbps modem. Furthermore, even if a fast connection device is

used, such as an ISDN, or Cable Modem, the Internet network itself is a shared medium and has limited bandwidth. Therefore, audio codecs are usually embedded to compress the voice data.

To guarantee the interoperability between IP voice devices from different vendors, the International Telecommunications Union (ITU) developed H.323 as the standard for telephony over IP network. H.323 defines common procedures for call setup, data compression, and data transport.

In a general sense, IP telephony can be thought of as providing a "virtual" point to point connection for voice services over Internet. An IP voice device is basically a gateway to connect the regular telephone system to the Internet. The following example demonstrates how a call would be placed. A user in Indianapolis wants to call a friend in Paris. He picks up his IP voice device handset (or activates a virtual handset on a computer screen for a "built-in" version) and hears a dial-tone like a regular telephone dial-tone. Then, he dials his friend's Paris phone number. The call travels over the Internet to some Switching Server provided by the IP telephony service provider. The Switching Server will connect the call to his friend's IP voice device and initiates the call. If his friend has only a regular telephone, the Switching Server will connect the call to a gateway in Paris. The gateway in Paris then initiates a call over the public switched telephone network (PSTN) to the local Paris number. The cost to make phone calls between Indianapolis and Paris using two IP voice devices is only the Internet access fee. If one party uses a regular telephone, the extra charge is merely that of a local dial call.

Depending on the Internet connection, there are at least two methods for making calls using an IP voice device: dial-up connections, and direct connections. With a dial-up connection, a user first calls an ISP (Internet service provider) over a regular dial-up line to set up an Internet connection. Then, he will use the IP voice device handset to dial the phone number of the person he is calling. The present applicants recognize one problem with this approach is that the recipient must be online waiting for the call. So, the sender may have to first call the recipient using a regular phone to make the appointment. With a direct connection, a user places a call using the IP voice device just as he does with the regular telephone. The direct connection indicates a permanent open channel to the Internet such as ISDN, or a cable access device. For a

dial-up connection call, a phone that has been called won't ring unless the Internet connection is already established for this phone. For a direct connection call, a phone would ring like a normal telephone.

The are many advantages to IP/Internet telephony. One such advantage is reduced cost as described above. A low bit rate audio codec embedded in the IP voice device enables voice calls over a 28.8 Kbps modem. For a small reduction in voice quality, a person's monthly phone bill will be greatly reduced. If IP voice device used together with a cable modem, the private service network plus high bandwidth of the cable modem will provide very good sound quality. Even if the voice quality provided by a IP/internet voice device is unsuitable for all phone communications, a IP/internet voice device may be useful as a second-line residential phone. Also, the H.323 standard supports several well defined conference modes and, therefore, IP voice device is able to be used for multi-point conference calls. A "Web" dial-in service is advantageous for technical or customer support lines because, for example, an Internet address of a company's IP voice device can be embedded in the company's Web page and customers can then call the company simply by "clicking on" that Internet address. The cost associated with toll-free ("800" number) telephone numbers will be reduced as a result.

In addition, MSOs (cable television system operators) have recently become interested in adding inexpensive telephony services using a combination of an MSO's private HFC (Hybrid Fiber Coax) network and the public Internet. Voice signals are converted to digital values and transported across the networks using various established and proposed Internet protocols as IP (internet protocol) packets.

However, there are also problems associated with existing IP/Internet telephony systems. For example, the above-described systems involve some combination of additional or revised POTS (Plain Old Telephone System) wiring, additional or revised cable network wiring, or additional network interface boxes. In addition, any connection which replaces a PSTN (public switched telephone network) service (such as reuse of the existing POTS wiring within the home to replace PSTN services with HFC telephony services) may be required to supply so called "life-line" services. Some of these options require professional installation which may be costly, time consuming, and inconvenient for the user.

SUMMARY OF THE INVENTION

The invention resides, in part, in recognition of the above-described problems and, in part, in providing a system and method for solving these problems. In particular, the inventors recognize that the described problems are solved by providing a voice call over an Internet connection by receiving a signal modulated in a first format and representing internet protocol data packets of the voice call; demodulating the signal modulated in the first format; modulating the demodulated signal into a second format; transmitting the signal modulated in the second format via a wireless connection to a receiving device; and demodulating the signal modulated in the second format in the receiving device to complete the voice call.

An aspect of the present invention involves providing an internet telephony system using a wireless connection such as via the unregulated 900 MHz cordless phone spectrum or other spectrum allocated for wireless communications to provide an RF link between an IP connection device, a network interface box or set-top box; and one or more wireless handsets. A processing/control element in the network interface box would run the required IP protocols to establish and manage call set-up and teardown (currently defined within ITU-T H.323), translate the digital voice signal between IP and the local RF link protocol, and provide the RF base station function for the handset(s). Each handset would incorporate the other end of the RF link, and A/D and D/A functions to convert the voice signal to and from digital packets, and potentially apply some compression algorithm to improve bandwidth utilization. In a handset design which does not incorporate enough processing power to perform the compression function, this function could potentially reside in the network interface box.

Another aspect of the present invention involves a mechanism to establish a wireless interface to a telephone handset through a settop box that is tied into a cable network such as a hybrid coaxial cable network.

Another aspect of the present invention involves using a standard protocol such as the Internet Protocol to maintain a digital connection into a cable network while using an RF link to transmit compressed voice/data information between a telephone device such as a telephone handset and an interface unit such as a settop box.

Aspects of the present invention also involve providing for eliminating the need to add wiring, such as POTS wiring, to accommodate one or more handsets, or alternatively eliminating the need to add multiple cable drops and adapters such as POTS/HFC adapters. A wireless feature in accordance with aspects of the invention provides for coupling a network interface box to an existing cable outlet and for adding handsets as required without installation of additional outlets. In addition, aspects of the invention provide for multi-line Internet phone calls without rewiring. Another aspect of the invention involves adding an analog trunk interface wherein an IP voice device can be connected to a PBX device for providing an Internet PBX. For example, a user could dial a prefix, such as "9" to make a regular outside phone call, or dial a different prefix, such as "8" to make an Internet phone call.

In accordance with another aspect of the present invention, an IP voice device or a set top box provides for connecting to external equipment, such as a PC or Workstation, and utilization of computation power of external devices for providing additional features such as IP FAX service or video conferencing.

BRIEF DESCRIPTION OF THE DRAWING

The invention may be better understood by referring to the accompanying drawing in which:

Figure 1 shows, in block diagram form, an embodiment of a system incorporating aspects of the invention; and

Figures 2 through 7 show, in block diagram form, embodiments of portions of the system shown in Figure 1.

Figure 8 is a flow chart illustrating a method of operation according to the principles of the present invention.

DETAILED DESCRIPTION

In Figure 1, a system constructed in accordance with aspects of the invention comprises a PSTN network and a cable network coupled to a cable modem termination system. The PSTN Network and/or Cable Network provide alternative paths for coupling the system shown to the Internet, e.g., to an Internet service provider (ISP). The cable modem termination system is coupled to a gateway, such as in a home

environment, that comprises a cable modem network interface and first and second codecs for coupling to a conventional wired telephone via a subscriber line interface unit and/or to a wireless telephone unit via an RF modem interface, respectively.

Data transmission between the various units shown in Figure 1 occurs as follows. Data transmission between the PSTN network and the cable modem termination system shown in Figure 1 (path 1 in Figure 1) may occur in 64 Kbps/voice line format or in T1 or higher hierarchy. Data in the cable network (e.g., path 2 between the cable network and the cable modem termination system in Figure 1, or path 2 between the cable modem termination system and the cable modem network interface unit in the gateway in Figure 1) may be carried over TCP/IP compressed at various rates or uncompressed linear at 64 Kbps/voice line. Data transmission between the cable modem network interface and the first codec (path 3 in Figure 1) may occur in linear PCM format at 64 Kbps/voice line. Data transmission between the cable modem network interface and the second codec (path 4 in Figure 1) may occur in linear format at 64 Kbps/voice line or compressed at various rates. Data communication on path 5 in Figure 1 (between the first codec and the subscriber line interface unit) may be in companded format at 64 Kbps/voice line. Data communication via path 6 in Figure 1 (between the second codec and the RF modem interface) may be in linear format at 64 Kbps/voice line or in compressed format at various rates. Data communicated to and from the subscriber line interface unit (path 7 in Figure 1) may occur in analog format (e.g., for an RJ11 connector) and data communicated to and from the RF modem interface (path 8 in Figure 1) may occur in RF digital modulation format.

In embodiments shown in Figs. 2 and 3, IP telephony compression algorithms, call setup, and a cordless telephone adapter are incorporated into an IP connection device or a client server. An example of an embodiment of such a device is a device referred to as a Network Computer (NC) which is a computer similar to a personal computer (PC) that is intended primarily for providing an interface to the Internet. That is, a network computer is intended primarily to provide computing power and features sufficient, for example, to connect to the internet, execute web browser software and provide email capability. A cordless telephone adapter in accordance with aspects of the invention would allow the convenience to call from any room in a house without expensive rewiring.

The phone would ring only when there is an incoming IP phone call, and would present dial tone, etc. when used to place a call.

Two exemplary embodiments of an IP connection device having a cordless phone interface are shown in Figures 2 and 3. The system shown in Figure 2 utilizes an analog cordless telephone interface such as CT-1 (46/49 MHz). The system shown in Figure 3 uitlizes a digital 900 MHz spread spectrum cordless telephone interface. The analog cordless IP voice device may provide a lower cost solution. However, a digital 900 MHz cordless IP voice device may be more advantageous in terms of voice quality and expandability. For example, a digital cordless phone typically provides better voice quality due to the noise cancelling capability of the digital system and a digital cordless IP voice device may have more than one handset. Also, a cordless IP voice device such as that shown in Figure 2 and/or 3 may be used for data service when used together with wireless modem.

The systems shown in Figures 2 and 3 may include a voice codec for compressing and decompressing the voice data if the modem of the IP connection device is running at low speeds. Table 1 lists some popular standard voice codec algorithms and their associated data rates.

Standard	Data Rate
G.711	64 kbps
G.723.1	5.3/6.3 kbps
G.728	16 kbps
G.729	8 kbps
GSM	13.3 kbps

Table 1 Voice compression standards

Figure 2 shows a client server device including IP voice features which, for example, may be included on an IP voice adapter card included in the client server device. The IP voice feature includes a CT-1 subsystem comprising RF transmitter circuitry Tx and receiver circuitry Rx, a programmable PLL synthesizer, a baseband (audio) processor, and a microprocessor interface. The components Tx and Rx and the PPL synthesizer are used to modulate and demodulate RF signals for transmission and reception of the wireless telephone signals. A duplexer is used to separate the transmit and receive paths

of the RF communications. As discussed before, a PCM codec integrated with filters may be needed to provide A/D and D/A conversions and compression, as well as the transmit and receive filtering of the signals. The digital signal processing (DSP) unit may be, for example, an integrated circuit (IC) that implements the voice codec under the control of the CPU (central processing unit) which may be a microprocessor. The CPU provides the central control of the wireless IP interface device shown in Fig. 2. The CPU is connected to the various components of the device via a data control bus. The CPU has a built in memory for storing the required control codes, including implementation of the H.323 standards and the TCP/UDP/IP protocols.

Figure 3 shows another exemplary IP connection device having a digital cordless phone interface such as a 900 Mhz interface. A baseband device usually includes a spread-spectrum modem, an audio engine (PCM, DTMF, etc.), a voice codec, and a microcontroller. The components Tx and Rx and the PLL synthesizer are used to modulate and demodulate RF signals for transmission and reception of the wireless telephone signals, to and from the wireless handset. A duplexer is used to separate the transmit and receive paths of the RF communications. A DSP unit is used to implement the voice codec under the control of the CPU. The CPU, or central processor, provides the central control of the wireless IP interface device as shown in Fig. 3. The CPU is connected to the various components of the device via a data control bus. The CPU has a built in memory for storing the required control codes, including impelmentation of the H.323 standards and the TCP/UDP/IP protocols.

Another aspect of the present invention is a wireless internet telephony system to be connected to a cable network. A network architecture in accordance with the principles of the present invention is shown in Figure 4.

In Figure 4, an interface to cable network(100) comprises a cable modem termination in the physical layer that has a bi-directional channel connected to the hybrid fiber coax network (105). The physical layer modulation scheme may comprise, for example, Quadrature Amplitude Modulation (QAM). The transport mechanism may comprise TCP/IP. In order to enable the voice application over the cable modem, the network interface unit may employ a protocol such as H.323 over TCP/IP. This enables signaling, call set up and other functions. The voice (fax and

analog modem is included in this paradigm) data itself may be carried in a compressed or uncompressed format. For example, companded voice data at 64 Kbps can be carried over the cable network embedded in TCP/IP packets. Alternatively, it may be compressed using one of many voice compression methods and carried over the cable network embedded in TCP/IP packets. Certain types of data cannot be compressed (example fax or analog modem) and need to be carried in a linear format.

Figure 4 also depicts a wireless interface (104) to a plurality of handsets or receiver devices (101, 102, 103...). The protocol between the base device (100) and the handsets may be entirely proprietary or some standard interface. Additionally, the data format or voice (compressed in one of many possible algorithms or uncompressed) may be different in the RF network as compared to the format in which the voice was carried over the HFC network.

Advantages associated with maintaining the same data format (e.g., compression scheme) in the wired (cable) and wireless network are:

1. only a single encoding/decoding process is necessary which, in a home environment, can take place at the wireless handset or mobile terminal (multiple transcoding processes normally result in degradation of the original source material); and
2. the base station (e.g., in the home) is transparent to the data from the handset or mobile terminal.

An advantage associated with maintaining different data formats (e.g., compression scheme) in the wired (cable) and wireless network is that certain compressed formats are specifically suited to be carried over certain transmission channels. Channel errors, depending on how they occur, can cause different degradation to the source material depending on the compression scheme that is employed. The wired and wireless environments are very different in terms of channel characteristics. Therefore, tailoring the coding scheme to match the characteristics of the channels may have some benefits in the overall system design.

Figure 5 shows further details of an exemplary embodiment of cable set-top box (100) in Figure 4. The cable channel (91) that carries both downstream and upstream data is usually frequency division multiplexed to enable simultaneous channels of operation. Further, within a specific channel. due to the nature of the shared cable medium.

multiple users may signal using a time division multiplexed access mechanism. This task is coordinated by the head end.

The cable interface (40) is a network interface unit (NIU) comprising of a modulator/demodulator pair and a processing unit for interpreting the incoming data stream and messages. One of the transport mechanisms employed is TCP/IP. The NIU receives data, demodulates decodes and extracts the information pertaining to specific voice channels in this application. It is also responsible for maintaining signaling information with the external network (for example using the H.323 protocol stack or any other commonly used signaling stack used in telephony). Additional features such as caller ID, messaging, voice mail etc. are features that are supported by the NIU. This is enabled by its interface with the Caller ID block (50), the external digital signal processor (10) with an embedded microprocessor (5) that coordinates the task of messaging, and voice compression/decompression as necessary.

The incoming messages are stored in compressed or uncompressed format in the message memory(60). Other system architectures may be used wherein the messages are stored in message memory in yet another compressed format to increase the time over which messages can be stored in a given amount of available memory. This task of additional compression/decompression may take place in DSP unit 10. The code memory (70) contains the code for the DSP engine. The RF cordless circuitry (20) is responsible for communicating with the handsets or mobile terminals and exchange specific information intended for each device. In addition to the exchange of data, 20 is also responsible for exchanging signaling and status information. The system shown in Figure 5 includes a common bus (80) between the functional components for data exchange, but a generalized architecture need not be limited to the bus structure shown in Figure 5. Additionally, messaging information and caller ID information are exchanged between 100 and the handsets or mobile terminals through the RF/cordless circuitry.

Figure 6 shows an exemplary embodiment of the receiver/set-top box 100 described earlier in regard to Figure 4 and referred to as unit 700 in regard to Figure 6. The transmit and receive signals into the cable network through the RF connector (796) are kept isolated using a diplexer (795). The cable tuner (705) and demodulator(710) convert the digitally modulated signal (for example QAM) into a composite digital bit stream which is delivered to a Medium Access Control – MAC (720) block

that performs the task of separating information into logical transport streams. Additionally, unit 720 is responsible for synchronizing with the cable head end in order to provide the settop box access control to the common cable medium for return channel information. The burst modulator (740) and power amplifier(730) create and send data in the return channel path back into the cable network.

The RF processing chain for processing the digital information from the cable network starts with the interface, or input/output (I/O) unit (760) which may be implemented as an application specific integrated circuit (ASIC) and which interfaces with a cordless phone processing unit (750), that also may be part of an ASIC or a separate ASIC, to create individual links with handsets or mobile receivers. Unit 750 is coupled to memory units DRAM 765 and ROM 770 for receiving stored processing instructions and for temporary data storage during processing. The data intended for each individual handset or mobile receiver may be time slotted, modulated and sent on the RF link through the RF connector (797). Additional information streams processed by the MAC processing block (720) may be directed to an ethernet port (783) through an ethernet controller (781) or an USB (Universal Serial Bus) port (784) through an USB controller (785) or an RS232 interface (791) through an RS232 driver(790).

The various functions shown in Figure 6 are connected to bus 721 for communication of data and control information between the functions and between the functions and CPU 786 which controls the operation of the functions in device 700. Also coupled to bus 721 are memory units 775 and 780 for storing control programs and data for CPU 786 and other functions in device 700. Power for unit 700 is provided by power supply 792. Also, while many of the processing blocks shown in Figure 6 may be optional depending on the specific application or product, the system shown in Figure 6 illustrates the composite nature of the data coming over the cable system. The path for the voice channels are of particular interest in regard to the present invention.

Fig. 7 shows a block diagram of an implementation of a wireless handset 101. The handset 101 comprises a DSP unit 201 including a microprocessor 210, a speaker, earpiece, RF circuitry and a keypad. The microprocessor 210 controls the various components of the wireless handset 101 via a system bus 202. The RF circuitry is connected to a RF antenna for transmitting and receiving RF wireless signals. A keypad 204

is used for an user to dial a phone number and for controlling other functions of the wireless phone. The DSP converts the analog signal into a digital signal to be transmitted over the RF spectrum if a digital transmission system is used. Memory 203 stores the program codes to be executed by the microprocessor 210.

It is to be understood that the embodiments and variations shown and described herein are illustrations only and that various modifications may be implemented by those skilled in the art without departing from the scope and spirit of the invention.

Figure 8 shows a flow chart illustrating a method of operation according to the principles of the present invention. At step 802, a signal is received by, for example, by the cable interface 40 of unit 100. This signal is demodulated by cable interface 40 into a demodulated signal at step 803. The unit may decide to further compress or decompress this demodulated signal under the control of the DSP unit 10 as described before. The DSP then causes this signal to be further modulated at step 805 by the RF/Cordless circuitry 20. This further modulated signal is then transmitted wirelessly to a wireless unit, for example, as shown in Fig. 7. After receiving this further modulated signal at the wireless unit, the wireless unit then demodulates this signal for completion of the IP voice call.

It is to be understood that the embodiments and variations shown and described herein are illustrations only and that various modifications may be implemented by those skilled in the art without departing from the scope and spirit of the invention.

CLAIMS

1. A method for processing a voice call over an internet, comprising the steps of:

receiving a signal modulated in a first format, said signal representing internet protocol data packets of said voice call;

demodulating said signal modulated in said first format;

modulating said demodulated signal into a second format;

transmitting wirelessly said signal modulated in said second format to a wireless device; and

demodulating said signal modulated in said second format in said wireless device.

2. The method of claim 1 further comprising the step of:
compressing said demodulated signal modulated in said first format before said transmitting step.

3. The method of claim 1 further comprising the step of:
decompressing said demodulated signal modulated in said first format before said transmitting step.

4. The method of claim 2 further comprising the step of: decompressing said demodulated signal modulated in said second format in said wireless device after said demodulating step in said wireless device.

5. The method of claim 1 wherein said first format is H.323 compliant.

6. The method of claim 1 wherein the first format comprises same modulation scheme as the second format.

7. The method of claim 1 wherein the first format comprises a different modulation scheme as the second format.

8. A system for processing a voice call over an internet, comprising:

means for receiving and demodulating a signal modulated in a first format, said signal representing internet protocol data packets of said voice call;

means for modulating said signal into a second format for wireless transmission to a wireless device; and

said wireless device further comprising means for demodulating said signal in said second format for completion of said voice call.

9. The system of claim 8 wherein said first format is H.323 compliant.

10. The system of claim 8 wherein said system further comprising means for compressing said demodulated signal modulated in said first format before said wireless transmission.

11. The system of claim 8 wherein said system further comprising means for decompressing said demodulated signal modulated in said first format before said wireless transmission.

12. The system of claim 10 wherein said system further comprising means for decompressing said compressed signal in said wireless device.

13. The system of claim 8 wherein the first format comprises same modulation scheme as the second format.

14. The method of claim 8 wherein the first format comprises a different modulation scheme as the second format.

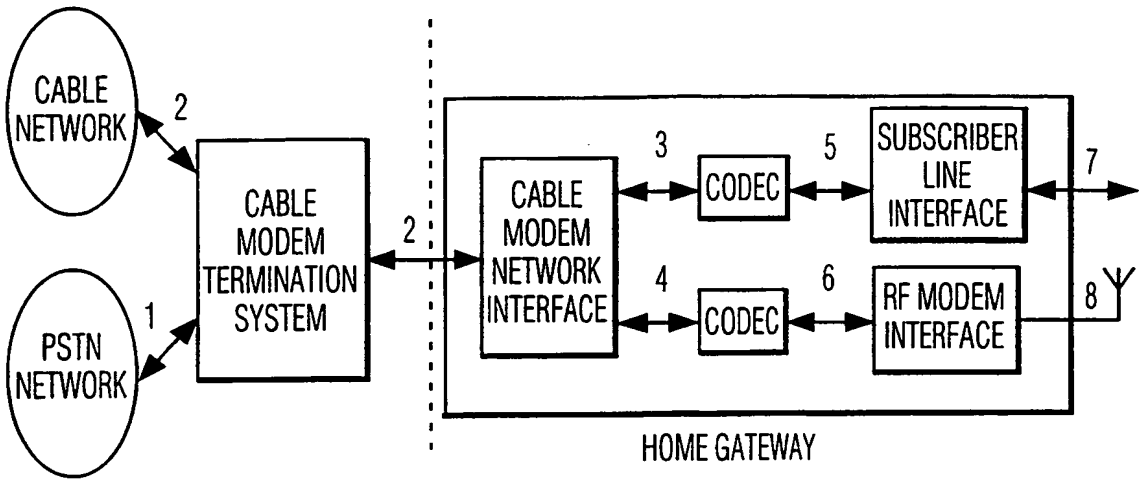
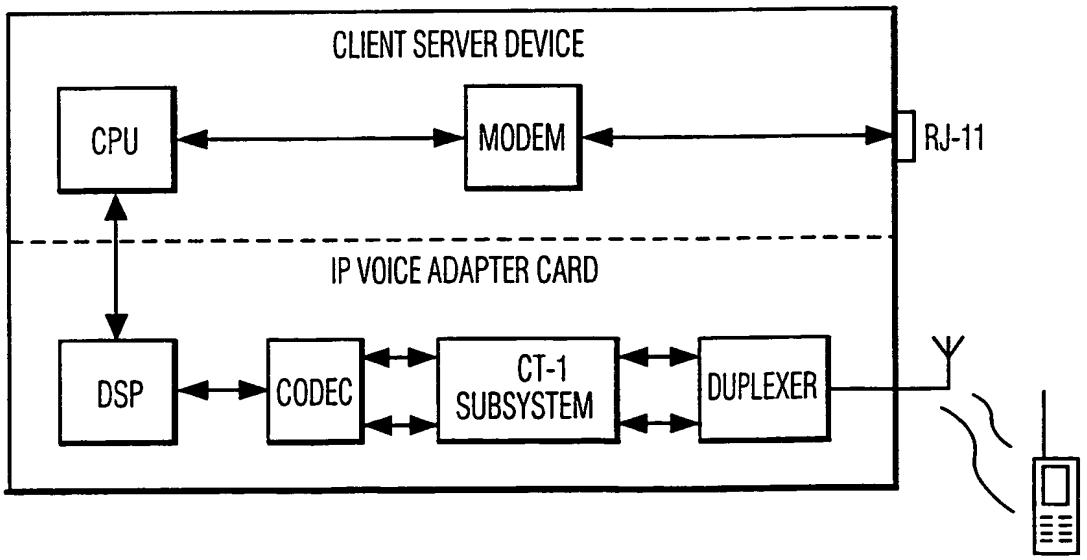

CABLE NETWORK
2
CABLE MODEM TERMINATION SYSTEM
2
PSTN NETWORK
1
CABLE MODEM NETWORK INTERFACE
3
CODEC
5
SUBSCRIBER LINE INTERFACE
7
4
CODEC
6
RF MODEM INTERFACE
8
HOME GATEWAY

FIG. 1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DSP
CODEC
CT-1 SUBSYSTEM
DUPLEXER

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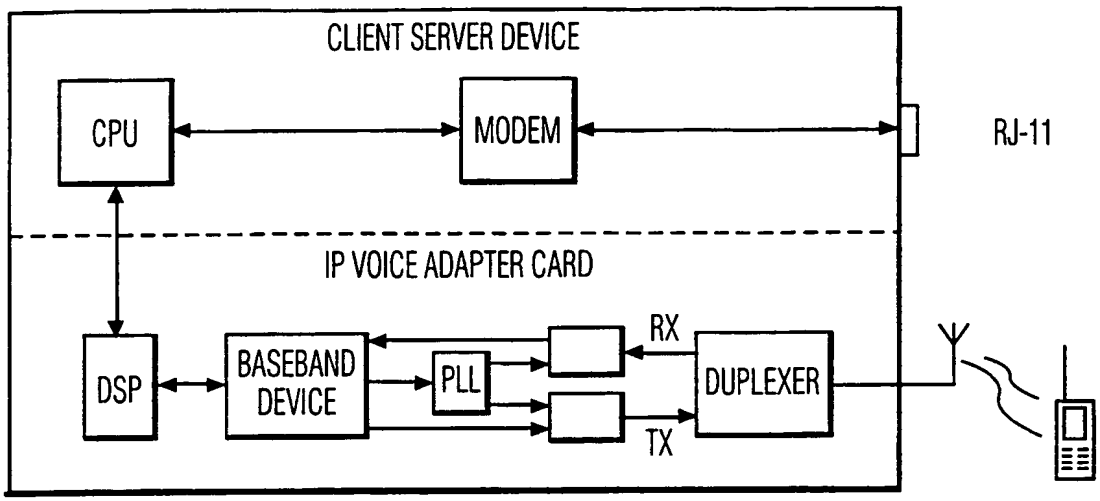
CLIENT SERVER DEVICE
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MODEM
RJ-11
IP VOICE ADAPTER CARD
DSP
BASEBAND DEVICE
PLL
RX
TX
DUPLEXER

FIG. 3

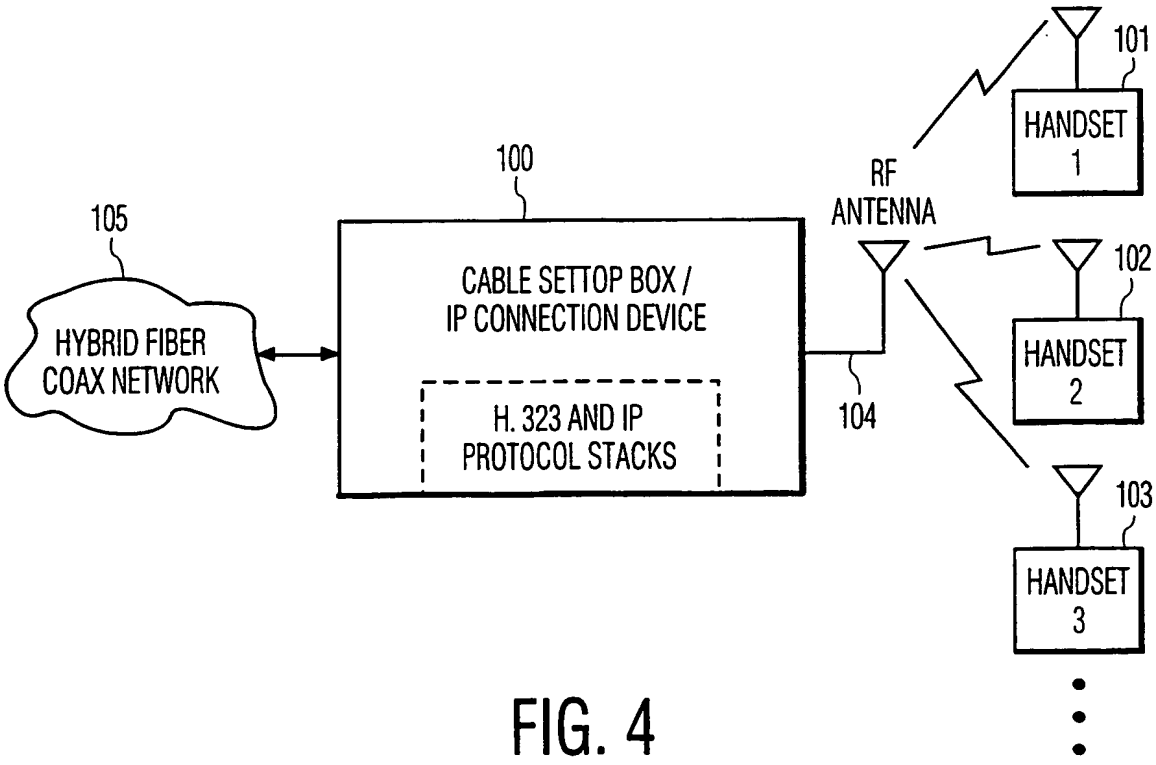
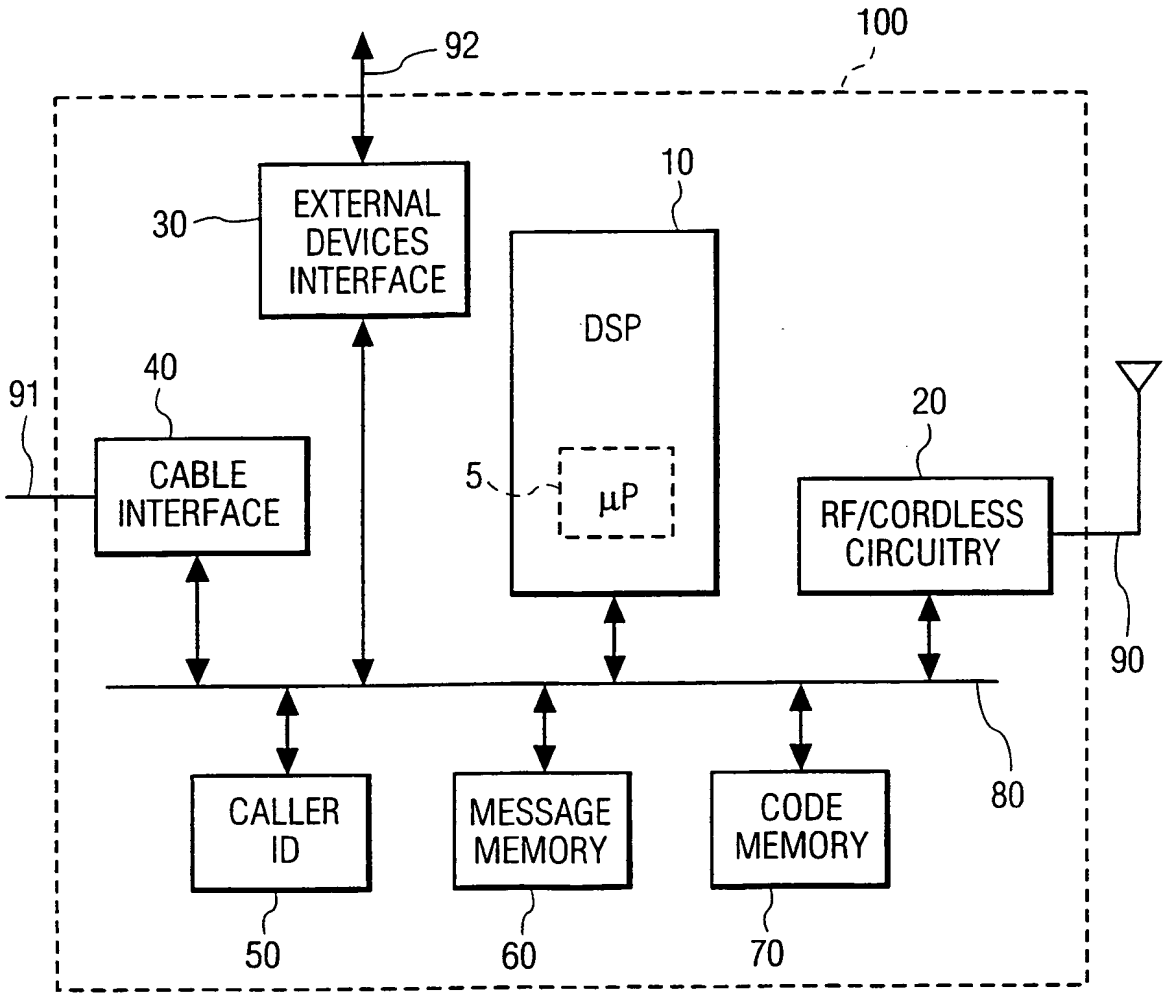

101
HANDSET 1
100
RF ANTENNA
105
CABLE SETTOP BOX / IP CONNECTION DEVICE
HYBRID FIBER COAX NETWORK
H. 323 AND IP PROTOCOL STACKS
104
102
HANDSET 2
103
HANDSET 3

FIG. 4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100
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DSP
40
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μP
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RF/CORDLESS CIRCUITRY
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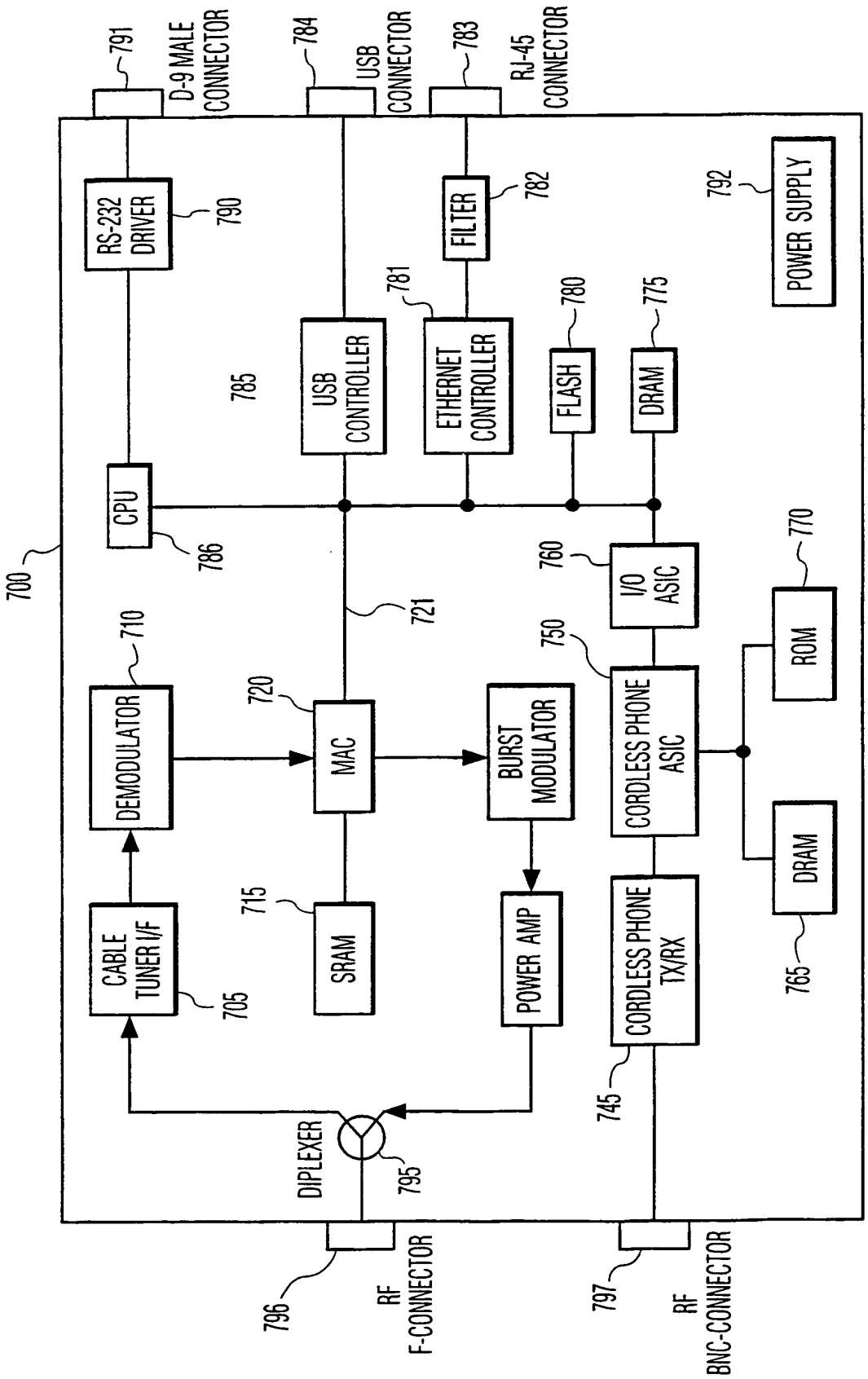
80
CALLER ID
MESSAGE MEMORY
CODE MEMORY
50
60
70

FIG. 5


791
D-9 MALE CONNECTOR
784
USB CONNECTOR
783
RJ-45 CONNECTOR
RS-232 DRIVER
790
FILTER
782
792
POWER SUPPLY
781
USB CONTROLLER
ETHERNET CONTROLLER
780
FLASH
775
DRAM
785
CPU
786
700
760
I/O ASIC
770
ROM
721
710
DEMODULATOR
720
MAC
BURST MODULATOR
750
CORDLESS PHONE ASIC
715
SRAM
POWER AMP
CABLE TUNER I/F
705
CORDLESS PHONE TX/RX
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DRAM
745
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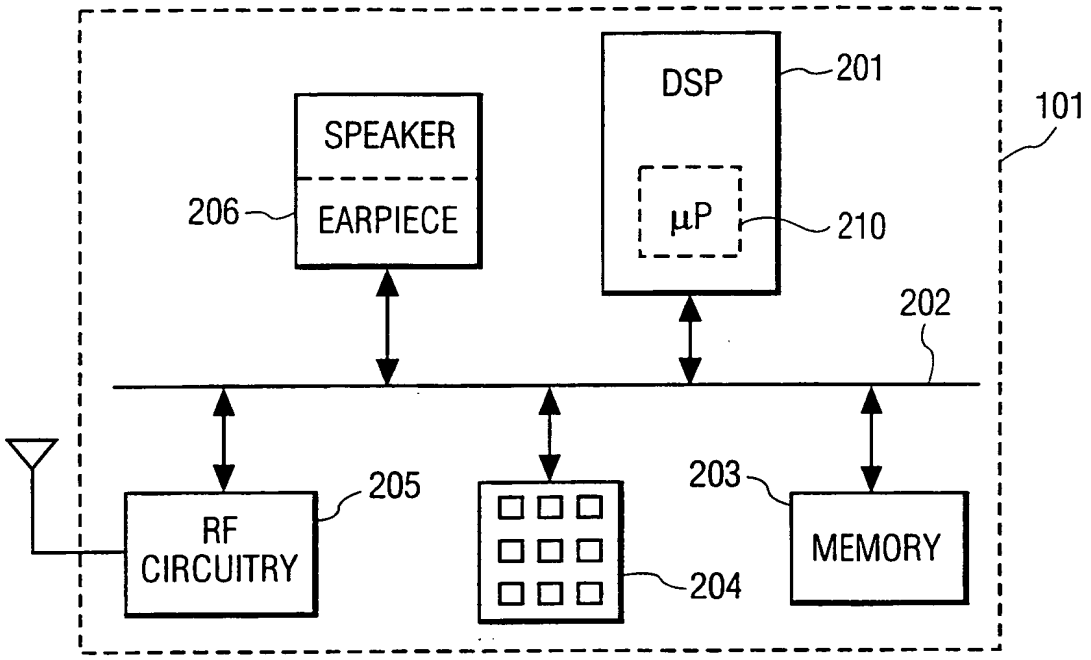
795
796
RF F-CONNECTOR
797
RF BNC-CONNECTOR

FIG. 6


DSP
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101
SPEAKER
206
EARPIECE
μP
210
202
205
RF
CIRCUITRY
203
MEMORY
204

FIG. 7

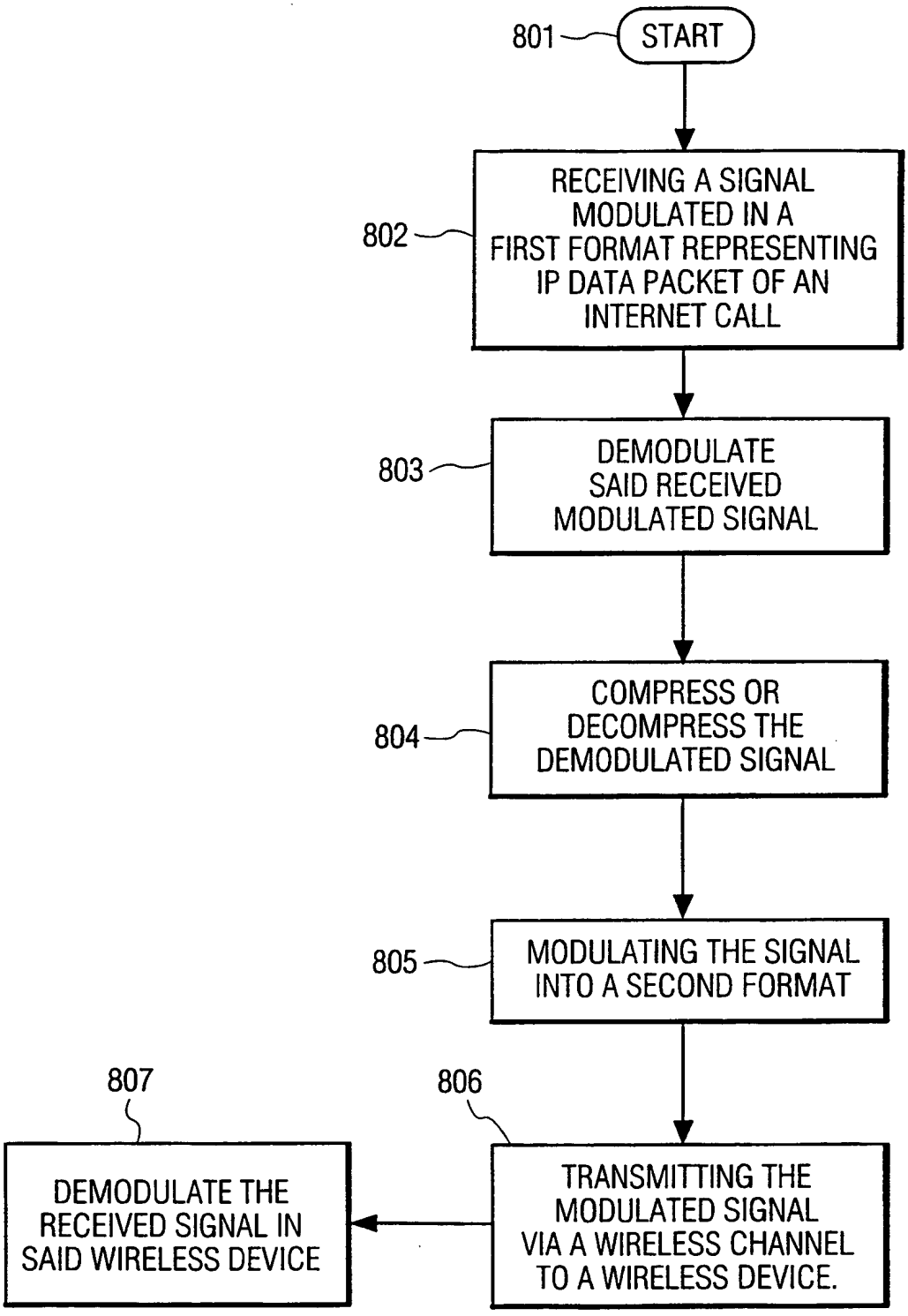
801
START
802
RECEIVING A SIGNAL MODULATED IN A FIRST FORMAT REPRESENTING IP DATA PACKET OF AN INTERNET CALL
803
DEMODULATE SAID RECEIVED MODULATED SIGNAL
804
COMPRESS OR DECOMPRESS THE DEMODULATED SIGNAL
805
MODULATING THE SIGNAL INTO A SECOND FORMAT
806
TRANSMITTING THE MODULATED SIGNAL VIA A WIRELESS CHANNEL TO A WIRELESS DEVICE.
807
DEMODULATE THE RECEIVED SIGNAL IN SAID WIRELESS DEVICE

FIG. 8

INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 99/01631

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H04M7/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H04M H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	WO 97 28628 A (LABS OF ADVANCED TECHNOLOGIES) 7 August 1997 see page 11, paragraph 1 - paragraph 3 see page 13, paragraph 3 see page 14, paragraph 1 - page 16, paragraph 4 see page 25, paragraph 3 - page 27, paragraph 2 ---	1-4,6-8, 10-14
A	WO 97 29581 A (LINK WORLDWIDE INC I) 14 August 1997 see abstract see page 15, line 18 - page 16, line 20 --- -/--	1-4,6-8, 10-14

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
11 May 1999	21/05/1999

Name and mailing address of the ISA	Authorized officer
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C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication,where appropriate, of the relevant passages	Relevant to claim No.
A	MOFFAT I G ET AL: "THE INTERNET TELEPHONE - A NEW PARADIGM" IEE COLLOQUIUM ON ADVANCES IN INTERACTIVE VOICE TECHNOLOGIES FOR TELECOMMUNICATION SERVICES,12 June 1997, pages 10/1-10/06, XP000197765 see the whole document ---	1-14
A	CHEN R -X ET AL: "AN I-PHONE SYSTEM DESIGN AND IMPLEMENTATION WITH A PORTABLE SPEECH CODING COPROCESSOR" IEEE TRANSACTIONS ON CONSUMER ELECTRONICS, vol. 43, no. 4, 1 November 1997, pages 1262-1269, XP000770882 see the whole document ---	1-14
P,X	WO 98 11703 A (SOLOMON YORAM ;SOLRAM ELECTRONICS LTD (IL)) 19 March 1998 see page 19, line 14 - page 20, line 6; figure 5 see page 34, line 4 - page 35, line 24 see figures 4,6-8,14A,B -----	1,6,8,13

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/US 99/01631

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO 9728628 A	07-08-1997	AU 1851397 A	22-08-1997
WO 9729581 A	14-08-1997	AU 2251497 A	28-08-1997
		CA 2245815 A	14-08-1997
		EP 0878086 A	18-11-1998
WO 9811703 A	19-03-1998	AU 3951797 A	02-04-1998

dial-up connection call, a phone that has been called won't ring unless the Internet connection is already established for this phone. For a direct connection call, a phone would ring like a normal telephone.

The are many advantages to IP/Internet telephony. One such advantage is reduced cost as described above. A low bit rate audio codec embedded in the IP voice device enables voice calls over a 28.8 Kbps modem. For a small reduction in voice quality, a person's monthly phone bill will be greatly reduced. If IP voice device used together with a cable modem, the private service network plus high bandwidth of the cable modem will provide very good sound quality. Even if the voice quality provided by a IP/internet voice device is unsuitable for all phone communications, a IP/internet voice device may be useful as a second-line residential phone. Also, the H.323 standard supports several well defined conference modes and, therefore, IP voice device is able to be used for multi-point conference calls. A "Web" dial-in service is advantageous for technical or customer support lines because, for example, an Internet address of a company's IP voice device can be embedded in the company's Web page and customers can then call the company simply by "clicking on" that Internet address. The cost associated with toll-free ("800" number) telephone numbers will be reduced as a result.

In addition, MSOs (cable television system operators) have recently become interested in adding inexpensive telephony services using a combination of an MSO's private HFC (Hybrid Fiber Coax) network and the public Internet. Voice signals are converted to digital values and transported across the networks using various established and proposed Internet protocols as IP (internet protocol) packets.

However, there are also problems associated with existing IP/Internet telephony systems. For example, the above-described systems involve some combination of additional or revised POTS (Plain Old Telephone System) wiring, additional or revised cable network wiring, or additional network interface boxes. In addition, any connection which replaces a PSTN (public switched telephone network) service (such as reuse of the existing POTS wiring within the home to replace PSTN services with HFC telephony services) may be required to supply so called "life-line" services. Some of these options require professional installation which may be costly, time consuming, and inconvenient for the user.

SUMMARY OF THE INVENTION

The invention resides, in part, in recognition of the above-described problems and, in part, in providing a system and method for solving these problems. In particular, the inventors recognize that the described problems are solved by providing a voice call over an Internet connection by receiving a signal modulated in a first format and representing internet protocol data packets of the voice call; demodulating the signal modulated in the first format; modulating the demodulated signal into a second format; transmitting the signal modulated in the second format via a wireless connection to a receiving device; and demodulating the signal modulated in the second format in the receiving device to complete the voice call.

An aspect of the present invention involves providing an internet telephony system using a wireless connection such as via the unregulated 900 MHz cordless phone spectrum or other spectrum allocated for wireless communications to provide an RF link between an IP connection device, a network interface box or set-top box; and one or more wireless handsets. A processing/control element in the network interface box would run the required IP protocols to establish and manage call set-up and teardown (currently defined within ITU-T H.323), translate the digital voice signal between IP and the local RF link protocol, and provide the RF base station function for the handset(s). Each handset would incorporate the other end of the RF link, and A/D and D/A functions to convert the voice signal to and from digital packets, and potentially apply some compression algorithm to improve bandwidth utilization. In a handset design which does not incorporate enough processing power to perform the compression function, this function could potentially reside in the network interface box.

Another aspect of the present invention involves a mechanism to establish a wireless interface to a telephone handset through a settop box that is tied into a cable network such as a hybrid coaxial cable network.

Another aspect of the present invention involves using a standard protocol such as the Internet Protocol to maintain a digital connection into a cable network while using an RF link to transmit compressed voice/data information between a telephone device such as a telephone handset and an interface unit such as a settop box.

CLAIMS

1. A method for processing a voice call over an internet, comprising the steps of:

receiving a signal modulated in a first format, said signal representing internet protocol data packets of said voice call;

demodulating said signal modulated in said first format;

modulating said demodulated signal into a second format;

transmitting wirelessly said signal modulated in said second format to a wireless device; and

demodulating said signal modulated in said second format in said wireless device.

2. The method of claim 1 further comprising the step of:
compressing said demodulated signal modulated in said first format before said transmitting step.

3. The method of claim 1 further comprising the step of:
decompressing said demodulated signal modulated in said first format before said transmitting step.

4. The method of claim 2 further comprising the step of: decompressing said demodulated signal modulated in said second format in said wireless device after said demodulating step in said wireless device.

5. The method of claim 1 wherein said first format is H.323 compliant.

6. The method of claim 1 wherein the first format comprises same modulation scheme as the second format.

7. The method of claim 1 wherein the first format comprises a different modulation scheme as the second format.

8. A system for processing a voice call over an internet, comprising:

means for receiving and demodulating a signal modulated in a first format, said signal representing internet protocol data packets of said voice call;

means for modulating said signal into a second format for wireless transmission to a wireless device; and

said wireless device further comprising means for demodulating said signal in said second format for completion of said voice call.

9. The system of claim 8 wherein said first format is H.323 compliant.

10. The system of claim 8 wherein said system further comprising means for compressing said demodulated signal modulated in said first format before said wireless transmission.

11. The system of claim 8 wherein said system further comprising means for decompressing said demodulated signal modulated in said first format before said wireless transmission.

12. The system of claim 10 wherein said system further comprising means for decompressing said compressed signal in said wireless device.

13. The system of claim 8 wherein the first format comprises same modulation scheme as the second format.

14. The method of claim 8 wherein the first format comprises a different modulation scheme as the second format.